PATENT APPLICATION
Attorney Doc. No. 2705-119

In re application of: Shmuel Shaffer, Joseph F. Khouri, Michael E. Knappe, and John F. Wakerly

Serial No. 09/702,196 Examiner: Oanh L. Duong

Confirmation No. 9840

Filed: October 30, 2000 Group Art Unit: 2155

For: **METHODS, DEVICES AND SOFTWARE FOR REDUNDANT TRANSMISSION OF VOICE DATA OVER A PACKET NETWORK CONNECTION ESTABLISHED ACCORDING TO AN UNRELIABLE COMMUNICATION PROTOCOL**

Commissioner for Patents
P.O. Box 1450
Alexandria, VA 22313-1450

DECLARATION OF JOSEPH F. KHOURI UNDER RULE 37 C.F.R. 1.131

I, Joseph F. Khouri, declare the following:

1. I am one of the co-inventors of the subject matter described in the present U.S. pending patent application entitled: METHODS, DEVICES AND SOFTWARE FOR REDUNDANT TRANSMISSION OF VOICE DATA OVER A PACKET NETWORK CONNECTION ESTABLISHED ACCORDING TO AN UNRELIABLE COMMUNICATION PROTOCOL, U.S. Serial No. 09/702,196, filed October 30, 2000.

2. I currently work for Cisco Systems, Inc. My work mailing address is 170 West Tasman Drive, San Jose, CA 95134-1706.

3. Before February 22, 2000, I and the other named inventors of U.S. Serial No. 09/702,196 conceived of the idea of a first device establishing a connection with a second device through a network according to a packet network communication protocol. The first device

transmitting to the second device original voice data in original packets through the connection and generating redundant data by replicating the original voice data and transmitting the redundant data to the second device.

4. Attached as Exhibit A is a disclosure document that was written prior to February 22, 2000, where I and the other named inventors of the '196 patent application describe our invention. The invention disclosure form was submitted to our employer, Cisco Systems. The Invention Disclosure Form describes how a modem transmission speed and error rate are monitored. Then based on that monitoring it is determined whether to transmit the same voiced packet only once or multiple times. The endpoint with the codec can signal to the remote side to multi-transmit packets or to signal to the nearest router to do that. The system as described in the invention disclosure can also monitor the available bandwidth and vary the number of voice packets retransmits according to the monitored available bandwidth.

5. The Ekudden et al. reference (US 2001/0041981 A1) was used to reject the claims of our application under 35 U.S.C. § 102(e) and 35 U.S.C. § 103(a). Applicants wish to "swear behind" these reference. Although Ekudden et al has an effective filing date of February 22, 2000 that precedes the application's effective filing date of October 30, 2000, applicants conceived of their invention prior to February 22, 2000 and were diligent in reducing the concept to practice up until the time the patent application was filed on October 30, 2000.

I, the undersigned, declare that all statements made herein of my own knowledge are true, and that all statements made on information and belief are believed to be true; and further, that these statements and the like so made are punishable by fine or imprisonment, or both, under Section 1001 of Title 18 of the United Stated Code, and that such willful false statements may jeopardize the validity of the application of any patent issuing thereon.

DATED this 2 day of MARCH, 2004.

Joseph F. Khouri

Joseph F. Khouri

Patent Idea Details for Idea #67801

GENERAL INFORMATION

Title: **A Method and Apparatus for Re-transmitting Voice Packets over Erroneous Modem Connections.**

ID: 67801

Patent No.: ---, ---

URL: [Application No. ---]

Inventors: Joseph Khouri (jkhouri), Michael Knappe (mknappe), Shmuel Shaffer (shaffers), and John Wakerly (wakerly)
More details on these inventors listed below.

Date Entered:

Date Modified:

Date Filed:

Date Issued: ---

Background: Modems usually operate at a variable speed, lower than the advertised maximum speed. For example, the 3COM 56K modems can operate at 56 Kbps top speed. Due to regulatory restrictions on the power output in the United States and Canada, a more realistic top speed is 53 Kbps. In real applications, speeds typically range from the 40s to the low 50s (Kbps), with the average in the mid- to upper 40s. Still there may be cases wherein, because of lower line quality, the performance is downgraded to the 30s.
During usual operation, the modem attempts to increase the transmission rate while monitoring the error rate. As the modem reaches the maximum speed feasible for a given connection, the error rate of the transmission link increases and the modem is programmed to throttle down the speed.
The ongoing attempt of the modem to maximize the transmission rate implies that the transmission errors are encountered and the higher application layers have to re-transmit the information. This scheme works well with TCP/IP data transmission but it fails when UDP voice packets are sent over modems with variable speed. As the modem attempts to increase its speed and hits the maximum feasible speed for a given link, transmission errors occur, voice packets are lost, and the perceived voice quality of the soft phone is degraded.
A similar problem is encountered when a wireless modem, e.g. Aironet, is used for transmitting voice packets from a soft phone. The wireless modem attempts to maximize its speed by ramping up the transmission speed until the transmission error exceeds a pre-determined threshold. Each time the error threshold is reached, voice packets are lost, and the voice quality is degraded.

This idea equally applies to any error prone transmission medium.

Summary: Our invention provides an extension to existing codecs transmitting voice packets over a modem connection (or any connections where transmission errors can occur).

Our invention consists of at least the following embodiments:

1) The new software monitors both the modem transmission speed and error rate, then based on that it will determine whether to transmit the same voice packet only once or multiple (i.e. n where n >= 2) times in a row. For example, if the Output of the codec is X and the effective transmit line speed is greater than 2X then the new software will double transmit the same voice packet when transmission errors are encountered.

2) The new software does not require a method to detect transmission errors and rather would be configured to re-transmit n times in a row depending on the available transmission bandwidth.

3) The new software will add a mechanism to signal to the remote side to multi-transmit packets (n times) or to signal to the nearest router to do that.

4) The new software can be extended to provide a way for applications (e.g. AVVID SoftPhone) to configure any of the following parameters as appropriate: a) the needed transmission throughput when packets are transmitted once (i.e. X), b) the error threshold to start the double/multiple retransmission (if applicable), and c) the maximum number of times to retransmit the same packet in a row (i.e. n).

5) The system may monitor the available bandwidth and vary the number of voice packet re-transmits (n). For example, when there is no Data to transmit, n (for voice packets) might be set to 3. Under low Data traffic n might be set to 2, and under high data traffic, n might be scaled down to 1.

The receiving end would off course need to detect a duplicate packet being received and would throw it away. Please note that UDP connections already do that.

Please note that in general, the number of retransmits in each direction might be different depending on the data traffic pattern.

While the idea of packet duplication is not new, we believe that this disclosure introduces the following new concepts:
1. Invocation of local packet duplication in response to a high (higher than a given threshold) link error-rate.
2. The ability to request a remote entity to start sending duplicate packets when packet loss in a receiver is detected.
3. Utilization of available bandwidth to enhance voice quality and reducing the number of transmitted packets to n=1 when data packets require the available bandwidth.

Advantages: The invention has the following advantages:
1. Improves voice quality for a soft phone or streaming voice media applications communicating over a wireless connection.
2. Improves voice quality for a soft phone or streaming voice media applications communicating over a dial-up connection.
3. The system adapts itself to the available bandwidth and thus it does not increase the latency for Data packets while attempting to improve voice quality.

Cisco Use: Soft phone is part of the AVVID application A2.0 release. The method described in this invention provides a mechanism for improving the voice quality of our soft phone when it is used over Variable Speed Modems or other communication devices where transmission errors can occur. This method makes the soft phone a viable solution for a traveling person who uses the soft phone from his hotel room over a dial up network and for the mobile user who is using a wireless modem. In addition, a SoftPhone user using any transmission medium where errors can occur (e.g. LAN, WAN, Intranet, Internet,...) would benefit from this invention.

Method of Detecting Use By Other Companies: Any company that offers a high quality voice connection over a dial up network or any network with transmission errors would infringe upon this patent. In addition, companies offering a high quality voice connection for mobile users over a wireless modem would infringe upon this patent.

Previous Public Use: ---

First Written Record Date: ---

First Written Record URLs: ---

Supporting Docs URLs: ---